REDUCED RANK ADAPTIVE FILTER

CROSS-REFERENCE TO RELATED APPLICATIONS

N/A

STATEMENT REGARDING FEDERALLY SPONSORED RESEARCH OR DEVELOPMENT

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FIELD OF THE INVENTION

The present invention is directed to a method of adaptive digital filtering and more particularly to such a method utilizing a reduced rank technique.

BACKGROUND OF THE INVENTION

Adaptive digital filters are known for use in various signal processing applications including speech and radar processing, adaptive beamforming, echo cancellation and equalization. These filters estimate the optimal filter coefficients from observed data, i.e. data representing the received signal to be processed.

Reduced rank filters estimate the filter coefficients with a relatively small amount of observed data. Various reduced rank techniques are known including Principal Components, Cross-Spectral and Partial Despreading methods. The former two methods require an explicit estimate of the signal subspace via an eigen-decomposition of

the input covariance matrix which is extremely complex. Although the latter technique is much simpler, it does not achieve near full rank performance when the filter rank D is significantly less than the full rank N.

Other known reduced rank techniques include a multi-stage Wiener filter as proposed by Goldstein et al. in "A Multistage Representation Of The Wiener Filter Based On Orthogonal Projections" IEEE Trans. Inform. Theory, 44 (7), November, 1998 and an adaptive interference suppression algorithm as proposed by Honig et al. in "Adaptive Reduced-Rank Residual Correlation Algorithms For DS-CDMA Interference Suppression" In Proc. 32 Asilomar Conf. Signals, Syst. Comput., November, 1998. These methods perform well and do not require an eigen-decomposition. The present invention is an improvement of these latter two methods.

BRIEF SUMMARY OF THE INVENTION

In accordance with the present invention, the disadvantages of prior adaptive digital filters noted above have been overcome. The reduced rank adaptive filter of the present invention projects a received signal onto a lower dimensional subspace where the reduced rank subspace is iteratively constructed by multiplying the last basis vector by the received sample covariance matrix. The reduced rank filter of the present invention achieves full rank performance with a projected signal subspace having a significantly smaller dimension than the dimension of the received signal subspace. Moreover, for interference suppression applications, the optimum filter rank does not substantially increase with the dimension of the received signal subspace. As such, the method of the present invention enables faster tracking and convergence with significantly less training samples than can be achieved with prior techniques.

More particularly, the method of the present invention filters successive, received signal samples to provide a desired signal wherein a group of N successive samples form a received sample vector of digital data having a N x 1 dimension. The method includes generating D + 1 basis vectors where D is less than N and represents the reduced rank dimension. Each successive basis vector is generated from a given or an estimated steering vector and successively greater powers of a covariance matrix for a sequence of received sample vectors of data. The first basis vector is formed from the steering vector. A reduced rank vector of digital data having a D x 1 dimension is generated from a matrix of D basis vectors and the received sample vector of data. D filter coefficients are generated from correlations between pairs of the basis vectors. The desired signal output from the filter is generated from the filter coefficients and the reduced rank vector of data.

The method of the present invention can be used in any adaptive filtering application including echo and noise cancellation, channel equalization, radar processing, adaptive antenna arrays and interference suppression. The method of the present invention is particularly advantageous for applications where long filter lengths are required and fast convergence and tracking are important. One such application is for space-time interference suppression in wireless Code-Division Multiple Access (CDMA) systems.

These and other advantages and novel features of the present invention, as well as details of an illustrated embodiment thereof, will be more fully understood from the following description and drawings.

BRIEF DESCRIPTION OF THE SEVERAL VIEWS OF THE DRAWING

Fig. 1 is a block diagram of the reduced rank adaptive filter of the present invention for a wireless CDMA communication system;

Fig. 2 is a flow chart illustrating the steps of the reduced rank adaptive digital filtering technique of Fig. 1;

Fig. 3 is a block diagram illustrating the generation of the received signal sample covariance matrix and steering vector;

Fig. 4 is a diagram illustrating the generation of the basis vectors; and

Fig. 5 is a flow chart illustrating the method of selecting the dimension of the reduced rank filter.

DETAILED DESCRIPTION OF THE INVENTION

The reduced rank adaptive digital filtering technique 10 of the present invention is illustrated in Fig. 1 for interference suppression in a wireless CDMA communication system having a receiver 12. Although illustrated for a wireless communication system, the filtering technique of the present invention can be used in any adaptive digital filter application. The receiver 12 picks up a received signal via an antenna 14 coupled to a radio frequency (RF) amplifier and mixer 16. The output of the amplifier/mixer 16 is coupled to an analog to digital converter 18 to provide a serial stream of baseband digital data samples representing the received signal. The output of the digital to analog converter 18 is coupled to a serial to parallel converter 20. The serial to parallel converter 20 takes N serial samples and outputs the N samples in parallel to provide a received sample vector $r(i)$ of digital data having a dimension of N x 1.

The received sample vector $r(i)$, which is referred to as an observation signal, is coupled to an adaptive digital filter 22, the output of which is an estimate of the desired signal $b(i)$, referred to as the approximate desired signal $\hat{b}(i)$. A generator 24 produces a matrix M of basis vectors with a N x D dimension where D is less than N and represents the reduced rank dimension. The generator 24 also produces

a vector of filter coefficients $\tilde{c}$ of dimension D x 1 from a sequence received sample vectors $r(i)$ for $i = 1, \ldots, B$ and from a sequence of the decision derived desired signals $b(i), i = 1, \ldots, B$ or a sequence of given desired signals or training symbols $b(i), i = 1, \ldots, B$. The given training symbols $b(i)$ initially used by the generator 24 may be stored, for example, in a receiver memory (not shown).

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The adaptive filter 22 generates a reduced rank or projected received signal vector $\tilde{r}(i)$ of dimension D x 1 by multiplying the Hermitian transpose of the matrix M by the received sample vector $r(i)$ as follows.

$$\tilde{r}(i) = M^{\dagger} r(i)$$

The adaptive filter 22 generates the approximate desired signal $\hat{b}(i)$ by multiplying the Hermitian transpose of the filter coefficient vector by the reduced rank vector as follows.

$$\hat{b}(i) = \tilde{c}^{\dagger} \tilde{r}(i)$$

The approximate desired signal $\hat{b}(i)$ is applied to a conventional slicer 23 that essentially rounds the approximate desired signal $\hat{b}(i)$ to a desired signal level $b(i)$ which may be used to generate the filter coefficients in a decision directed training mode.

The generator 24 generates D + 1 basis vectors where each successive basis vector is generated from a given or an estimated steering vector p and successively greater powers of a sample covariance matrix R for a sequence of B received sample vectors $r(i), i = 1, \ldots, B$. The first basis vector is formed from the steering vector p as described in detail below with reference to Fig. 4. The generator 24 generates a D x 1 vector of filter coefficients $\tilde{c}$ from a matrix and vector of correlations between pairs of generated basis vectors.

The generator 24 and adaptive digital filter 22 operate in accordance with the method depicted in Fig. 2. At block 26, the generator 24 determines the sample covariance matrix R for the sequence of received sample vectors $r(i), i = 1, \ldots, B$ and the steering vector p from the sequence of received sample vectors $r(i), i = 1, \ldots, B$, and the sequence of training/desired signals $b(i), i = 1, \ldots B$ in accordance with the block diagram of Fig. 3. Specifically, at block 28, the Hermitian transpose of the received sample vector is formed and at 30 is multiplied by the received sample vector and a weighting factor of 1 - w. The resulting product is added at 32 to the product formed at 34 by multiplying the weighting factor w by the previous covariance matrix R_{i-1} so as to form the sample covariance matrix R_i for the sequence of received sample vectors $r(1), \ldots, r(i)$. At 36, the conjugate of the given or desired signal $b(i)$ is formed and at 38 is multiplied by the received sample vector $r(i)$ and the weighting factor $1 - w$. The resulting product is added at 40 to the product of the previous steering vector p_{i-1} and the weighting factor w to form the new steering vector p_i. The steering vector p_i generated from the sequence of given training symbols $b(i), i = 1, \ldots, B$ or from the sequence of desired filter outputs $b(i), i = 1, \ldots, B$ as shown in Fig. 3 is an example of an "estimated" steering vector. It should be appreciated that Fig. 3 is only one example of a technique for generating the estimated steering vector. Other known techniques for generating an estimated steering vector can be used as well. A "given" steering vector can be used in accordance with the present invention as well. A "given" steering vector is a steering vector that is already known to the receiver and may be stored, for example, in a memory or buffer of the receiver.

At block 44, the generator 24 generates D + 1 basis vectors v_0 through v_D. The basis vectors are generated in accordance with the diagram depicted in Fig. 4. The first or initial basis vector v_0 is set equal to the steering vector p. Each successive basis vector is formed by multiplying the immediately preceding basis vector by the covariance matrix R so that the basis vectors are generated as follows.

$$v_o = p,\ v_1 = Rp,\ v_2 = R^2 p, \ldots, v_{D-1} = R^{D-1} p,\ v_D = R^D p$$

The generator 24 at block 46 forms the matrix M of basis vectors from the basis vectors v_0 through v_{D-1} so that the matrix M has a N x D dimension. Thereafter, the generator 24 determines the correlation scalars γ_1 through γ_{2D-1} from the basis vectors v_0 through v_D determined at block 44. The correlation scalar γ_{i+j} is generated by multiplying the Hermitian transpose of the basis vector v_i by the basis vector v_j as follows.

$$\gamma_1 = v_0^{\dagger} v_1,\ \gamma_2 = v_0^{\dagger} v_2 \ldots \gamma_D = v_0^{\dagger} v_D,\ \gamma_{D+1} = v_D^{\dagger} v_1,\ \gamma_{D+2} = v_D^{\dagger} v_2 \ldots \gamma_{2D-1} = v_D^{\dagger} v_{D-1}$$

Also $\gamma_0 = v_o^{\dagger} v_0$ and v_0 is assumed to be normalized to that $\gamma_0 = 1$.

The generator 24 at block 54 forms a correlation matrix Γ from the correlation scalars γ_1 through γ_{2D-1} as follows.

$$\Gamma = \begin{bmatrix} \gamma_1 & \gamma_2 \cdots \gamma_D \\ \gamma_2 & \gamma_3 \cdots \gamma_{D+1} \\ \vdots & \vdots \\ \gamma_D & \gamma_{D+1} \cdots \gamma_{2D-1} \end{bmatrix}$$

At block 54 the generator 24 also generates a reduced rank correlation vector y which is a D x 1 vector formed of the correlation scalars γ_0 through γ_{D-1} as follows.

$$y = \begin{bmatrix} \gamma_0 \\ \gamma_1 \\ \vdots \\ \gamma_{D-1} \end{bmatrix}$$

The generator 24 at block 56 then generates the filter coefficient vector $\tilde{c}$ which is a D x 1 vector generated by solving the following set of linear equations.

$$\Gamma \tilde{c} = y$$

The matrix M of basis vectors formed at block 46 and the filter coefficients $\tilde{c}$ generated at block 56 are applied to the adaptive digital filter 22 so as to generate the reduced rank vector $\tilde{r}(i)$ and the approximate desired signal $\hat{b}(i)$ output from the filter 22 in accordance with blocks 58 and 60. In particular, at block 58, the adaptive digital filter 22 generates the reduced rank vector $\tilde{r}(i)$ by multiplying the Hermitian transpose of the matrix M of basis vectors by the received sample vector of digital data $r(i)$ as follows.

$$\tilde{r}(i) = M^{\dagger} r(i)$$

At block 60, the filter 22 generates the approximate desired signal $\hat{b}(i)$ by multiplying the Hermitian transpose of the D x 1 filter coefficient vector $\tilde{c}$ by the reduced rank vector $\tilde{r}(i)$ as follows.

$$\hat{b}(i) = \tilde{c}^{\dagger} \tilde{r}(i)$$

It has been found that the reduced rank adaptive digital filter described above achieves near full rank performance with a value of D equal to 8 or less for a Direct Sequence (DS) Code Division Multiple Access (CDMA) communication system. For other applications, the value of D can be determined in accordance with the method depicted in Fig. 5. As shown in Fig. 5, at block 60 an index n is initialized to 1. Thereafter, at block 62 the n^{th} basis vector v_n is generated by

multiplying the previous basis vector v_{n-1} by the covariance matrix R where the first basis vector is the steering vector p so that

$$v_n = R^n p$$

as discussed above. Thereafter, a matrix M_{n-1} of basis vectors is formed at block 64 as follows.

$$M_{n-1} = [v_0, v_1, \dots v_{n-1}]$$

At block 66, the orthogonal projection u_n, which is a N x 1 vector, is generated as follows.

$$u_n = v_n - M_{n-1}(M^{\dagger}_{n-1} M_{n-1})^{-1} M_{n-1}^{\dagger} v_n$$

The respective lengths l and g of the vectors u_n and v_n are computed at block 67 in accordance with the following equations.

$$l = \|u_n\|$$

$$g = \|v_n\|$$

where l and g are scalars. Thereafter, at block 68, it is determined whether the length l divided by the length g is greater than a small constant δ, where δ may be set equal to 0.01 for example. If so, n is incremented by 1 and steps 62 through 68 are repeated. Steps 62 through 70 are repeated until l divided by g is not greater than δ for a given value of n. At that point, D is set equal to n - 1.

In an alternative method, the value of D is chosen to minimize the "*a posteriori* Least Squares cost function C" where

$$C = \sum_{m=m_0}^{i} \left| b(m) - \tilde{c}^{\dagger}{}_D \tilde{r}(m) \right|^2$$

where $\tilde{c}_D$ is the reduced rank filter coefficient vector generated from the sequence of $b(1), \dots b(B)$ and the sequence of $r(1), \dots r(B)$ and $B < m_o$.

The reduced rank adaptive filter of the present invention projects the N x 1 received signal vector $r(i)$ onto the lower dimensional

subspace which results in a D x 1 vector $\tilde{r}(i)$ where D can be much less than N while still allowing the filter 10 to achieve near full rank performance. For interference suppression applications, the optimum filter rank D does not substantially increase with the dimension of the signal subspace N. This enables much faster tracking and convergence than can be achieved with prior techniques. Furthermore, the method of the present invention does not rely on an estimate of the signal space via, for example, eigen-decomposition so that the complexity of the present method is substantially reduced. Although the method was described in detail for a wireless communication system application, it can be used in any adaptive filtering application including, echo and noise cancellation, channel equalization, radar processing, adaptive antennae arrays, interference suppression, etc.

Many modifications and variations of the present invention are possible in light of the above teachings. Thus, it is to be understood that, within the scope of the appended claims, the invention may be practiced otherwise than as described hereinabove.

What is claimed and desired to be secured by Letters Patent is: